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A Sound Level Distribution Model for Symphony Orchestras: Possibilities and Limitations

Remy Wenmaekers and Constant Hak
Eindhoven University of Technology

Musicians in a symphony orchestra rely on the direct and reflected sound on a concert hall stage to be able to hear each other. Besides ensemble conditions, members and directors of symphony orchestras are concerned about the noise levels musicians are exposed to. However, the actual contribution of the different parts of the sound field cannot be derived from sound level measurements in orchestras. In this article, a prediction model is presented that can be used to investigate the distribution of the direct, early reflected, and late reflected sound from all musicians to the total sound level at a single musician's position. It is shown that the contributions of each different aspect to the total sound level are in the same order of magnitude. In some cases, the direct sound dominates, while in other cases, the early or late reflected sound does. Considerable variations in sound levels are found between a concert hall, rehearsal room, and orchestra pit, due to the difference in room acoustical properties. An example is presented of calculated sound levels for a violin's position in the orchestra for the 3 halls. The results from the example show that the model has potential for studying the influence of architectural as well as acoustical aspects on the sound levels that occur in a symphonic orchestra, both from a health and musical point of view.

Keywords: orchestra, sound level, acoustics, support, music

Sound levels and loudness play an important role in the performance of and listening to music (Beranek, 2011). The conductor and musicians in an orchestra search for the appropriate sound level balance between different instruments. This good balance is not only important for the listeners in the audience area but also crucial for the musicians themselves to be able play together easily (Gade, 1989), for instance, to control timing (Nowicki, Prinz, Grosjean, Repp, & Keller, 2013). For more information about ensemble playing and stage acoustics, an extensive literature review can be found in Gade (2011).

Besides ensemble conditions, members and directors of symphony orchestras are concerned about the noise levels musicians are exposed to. In accordance with European Directive, 2003/10/EC (Directive, 2003), professional musicians should be protected from noise levels that may cause hearing damage. The results of earlier investigations has shown that the noise levels within an orchestra can cause hearing loss (Jansen, 2009). Also, research has shown that the noise level will vary between different musicians playing different instruments and musical pieces, and between musicians being positioned differently on various stages (Schmidt, 2011).

Factors that determine the sound levels can be musically based: the piece and its interpretation by the conductor and musicians; or acoustically based: the impact of the stage and hall reflections. Schärer Kalkandjiev and Weinzierl (2013) have shown in a case study that a highly experienced solo cello player responded to a louder acoustical environment (higher Sound Strength or G) by reducing his power, while a lack of reverberation (not necessarily a lower G) encouraged the musician to play more powerful. In terms of sound exposure, this results in an interesting contradiction: by playing in a louder and more reverberant environment, the output power is reduced by the musician. What if the own instruments' direct sound is causing most of the exposure? Then, the total exposure might be lower in a louder acoustic environment.

To further investigate the balance between direct sound and reflected sound, in this article, we will focus on the acoustical aspects of sound levels. Sound reflected from the stage boundaries arrives relatively early after the direct sound. The (much) later reflected sound is mostly caused by reflecting surfaces in the audience area. To some extent, the amount of early and late reflected sound can be controlled separately by the individual design of the stage and design of the hall. Therefore, it is interesting to study the separate contributions of direct, early and late sound to the total sound exposure of the musicians.

The contribution of each instrument and each acoustical aspect to the total sound level cannot be separately determined from sound (pressure) level measurements (Rodrigues, Freitas, Neves, & Silva, 2014). As a solution to this problem, a sound level prediction model is presented in this article in which the direct, early reflected, and late reflected sound energy transfer is calcu-

Remy Wenmaekers and Constant Hak, Department of the Built Environment, Eindhoven University of Technology.

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Correspondence concerning this article should be addressed to Remy Wenmaekers, Department of the Built Environment, Unit BPS, Laboratorium voor Akoestiek, Eindhoven University of Technology, P.O. Box 513, 5600 MB Eindhoven, The Netherlands. E-mail: r.h.c.wenmaekers@tue.nl

lated using the sound power and directivity of symphony orchestra instruments together with stage acoustic parameters Early and Late Support. The goal of the model is to study trends of the contributions of the different instruments and different acoustical aspects to the total sound level received by every musician in the orchestra. The possibilities and limitations of the model will be discussed, and first results will be presented based on state of the art knowledge. However, the trends calculated by the current model should be compared with subjectively critical limits (however scarcely available) while considering the uncertainties and limitations of the model. We hope that our findings, derived from an architectural acoustics point of view, may inspire other fields of research concerning the understanding of sound exposure and of musical performance.

In this article, the possibilities of the model will be described in the "Model" section, together with discussions of its limitations. In the "Input and Output" section after that, an example of the necessary input data will be presented. In the "Validation" section, a study is presented to validate (part of) the model. Then, a case study is presented that show the possibilities of the model to give insights in the sound levels within a symphony orchestra, both from a health as well as from a musical point of view. This article concludes with a discussion on further development of the model and a conclusion.

Model

A visual representation of the sound level prediction model, together with the relevant parameters, is presented in Figure 1. The various parameters will be explained in the following paragraphs.

Sound Path

The transfer of sound from a sound source to a receiver in a room can be fully described by the room impulse response, which can either be measured or predicted. In this model, the impulse response is divided into three typical room acoustical aspects to study the balance between them: the direct sound, the early reflected sound, and the late reflected sound. The direct sound path is of interest to study the influence of distance, instrument directivity, and the attenuation of sound by objects. The early reflected sound is generally considered to be meaningful for ensemble

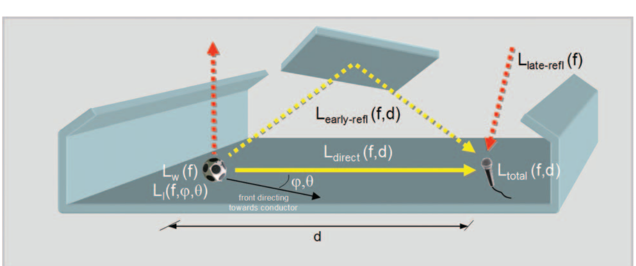

Figure 1. Summary of the sound level prediction model from one source to one receiver on a concert hall stage. The direct sound level, L_{direct}, is calculated based on measured instrument directivity and distance. The early and late reflected sound level, $L_{early\text{-}refl}$ and $L_{late\text{-}refl}$, respectively, are calculated based on measured room acoustic parameters. See the online article for the color version of this figure.

playing on stage while the late reflected sound may contribute to a sense of feedback from the hall. However, late reflected sound can also be detrimental for ensemble playing, as it can mask direct and early reflected sound (Gade, 2011). The direct sound is calculated analytically using the directivity of the sound source and receiver, together with an empirical model to include the effect of the obstruction by other orchestra members into the model. The early and late reflected sound energy is estimated from measured room impulse responses using an omnidirectional sound source and receiver on an empty stage.

In this article, available measured data from empty stages will be used. However, the impulse response should ideally be determined under the same conditions as a concert or rehearsal, which implies that the orchestra and/or audience should be taken into account. It is virtually impossible to meet such conditions during a measurement, as it would demand using transducers with the exact directivity as musical instruments and having orchestra (or even audience) present while performing measurements for hours. Results from modeled impulse responses might be used instead. This would be a future possible solution to include source directivity in the model calculations for the reflected sound. However, so far, no validated method exists to model the influence of orchestra members as obstructing objects in a geometrical acoustical model.

Sound Source Directivity

In general, a sound source can be described by the sound intensity $L_I(f, \varphi, \theta, d)$ with independent parameters frequency (f), orientation (elevation φ and azimuth θ), and distance (d). A musical instrument cannot easily be defined by these parameters, because the spectrum and directivity may change per note and playing style. When assessing sound levels, however, one is often interested in an average value over time. It may then be legitimate to use time average values for a musical instrument's directivity. In this model, measured average values of sound intensity per angle $L_I(f, \varphi, \theta)$, for common orchestral instruments at free field distance, are used from Pätynen & Lokki (2010), obtained from averaging over several tones within the instruments' tonal range.

The relative sound intensity $L_I(f, \varphi, \theta)$ of the instruments is made available by Pätynen & Lokki (2010), for the octave bands 125 Hz to 16 kHz, in accordance with the common loudspeaker format (CLF) with steps of 10 degrees. The CLF format is defined by 36 arcs running from the front to the back of the sound source. This results in a finer grid in the front direction where the directivity of a loudspeaker is usually the highest. In contrast, in this model, the CLF output is calculated without rotating the base of the original coordinate system 90 degrees downward on the transverse axis (Pätynen, 2009), see Figure 2. This way, the angle between a source and receiver is easily defined by elevation φ and azimuth θ. This will reduce the fineness of the grid in front of the sound source; however, as the CLF data points have been interpolated from only 20 microphone measurements, the information is still an interpolation from an even more coarse grid. The front orientation of the directivity is defined as the frontal viewing direction of the musician.

Direct Sound of Other Musicians' Instruments

The direct sound path depends on the source-to-receiver distance and orientation of the source relative to the receiver, assum-

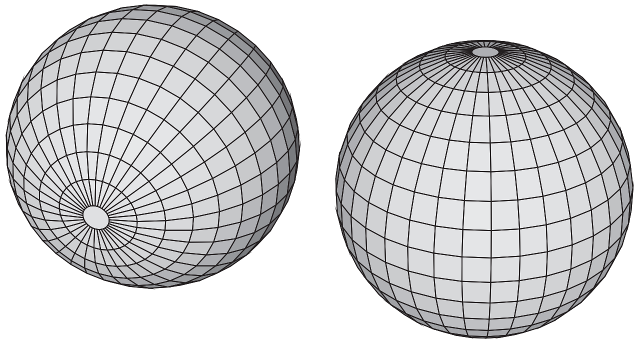

Figure 2. The common loudspeaker format is defined by 36 arcs running from the front to the back of the sound source, see left figure. In this model the common loudspeaker format output is calculated without rotating the base of the original coordinate system 90 degrees downward on the transverse axis, see right figure. This way, the angle between a source and receiver is easily defined by elevation φ and azimuth θ.

ing that both musicians face the conductor. Besides that, the attenuation of the orchestra as obstruction between the source and receiver is included from a model based on measured values of $\Delta L(f)$ by Dammerud and Barron (2010). In terms of assessing sound exposure in accordance with ISO 9612, a measurement should be done positioning an omnidirectional microphone on the shoulder at the most exposed ear. However, Schmidt (2011) concluded that positioning small omnidirectional microphones in front of the musicians' ear canal is a more appropriate method to measure sound exposure of musicians, because it better represents the exposure from the own musical instrument (for instance, it is often not possible to put a microphone on a violin player's left shoulder) and it is not always possible to predict the most exposed ear. In the model presented in this article, a similar method is used in which the sound level is calculated for the two ears while taking into account the transfer function of the receiver's head (HRTF). The HRTFs were measured for the 0-degree elevation while rotating in 10-degree steps in an anechoic room using omnidirectional DPA 4060 miniature condenser microphones in front of the ear canal of a B&K Type 4128C head and torso simulator, see Figure 3. The HRTFs were calibrated using the Sound Pressure Level (SPL) of the same microphones in the free field at center position of the head.

L_{direct} is calculated as

$$L_{direct}(f, d) = L_{eq;1m}(f, \phi, \theta) - 20\,\mathrm{lg}(d) + \Delta L_{orch}(f, d) + \Delta L_{ear}(f, \theta) \tag{1}$$

and

$$\Delta L_{orch}(f, d) = a(f) \times d + c(f) \tag{2}$$

where $L_{eq;1m}(f, \varphi, \theta)$ is the sound level in dB at 1 m distance per octave band 125 to 8000 Hz at elevation φ and azimuth θ in degrees estimated from measured values of sound intensity $L_I(f, \varphi, \theta)$ and $L_{eq;1m;front}(f)$ derived from the frontal anechoic recordings of every instrument; d is the source receiver distance in meters; $\Delta L_{orch}(f, d)$ is the attenuation by the orchestra in dB estimated from scale model measurements using an attenuation factor "a" in dB loss per meter through the orchestra and a constant "c" in dB for the overall shift of attenuation due to the effect of the floor and orchestra reflections, see Table 2 in Dammerud and Barron (2010), flat floor path A; and $\Delta L_{ear}(f, \theta)$ is the HRTF expressed as a difference in level between an omnidirectional microphone in front of the ear canal and the same microphone in the free field at the center head's position.

Direct Sound of the Own Instrument

The direct sound level of the own instrument also needs to be modeled. Pätynen & Lokki, (2010) determined the sound intensity per angle $L_I(f, \varphi, \theta)$ per instrument using the musicians head in the center. In our model, however, a reference point on the musical instrument itself needs to be regarded as the "point source" for the direct sound of the own instrument (even though this "point" can vary depending on notes played and playing style). The chosen reference points are explained in the "Validation" section. Hereby, the actual distance between the instrument and the reference microphone is taken into account, see Figure 4. For each individual ear, the direct sound of the own instrument is calculated as

$$L_{direct;own}(f, d) = L_{eq;microphone}(f, \phi, \theta) - 20\,\mathrm{lg}\left(\frac{d_{instrument\_to\_ear}}{d_{microphone\_to\_instrument}(\phi, \theta)}\right) \tag{3}$$

where $L_{eq,microphone}$ is the sound level at the microphone on a straight line crossing the ear and the reference point on the musical instrument; $d_{instrument\_to\_ear}$ is the distance between the reference point on the musical instrument and the ear; and $d_{microphone\_to\_instrument}$ is the distance between the microphone position and the reference point on the musical instrument.

The musicians' ear is in the near field of the musical instrument within less than 1 m distance and the far-field rules, like the inverse square law, might not be sufficiently reliable. Therefore, a validation study has been performed based on new measurements in an anechoic room with several musicians, which will be presented in the "Validation" section of this article. Note that the HRTF is not included in the prediction of the own instruments'

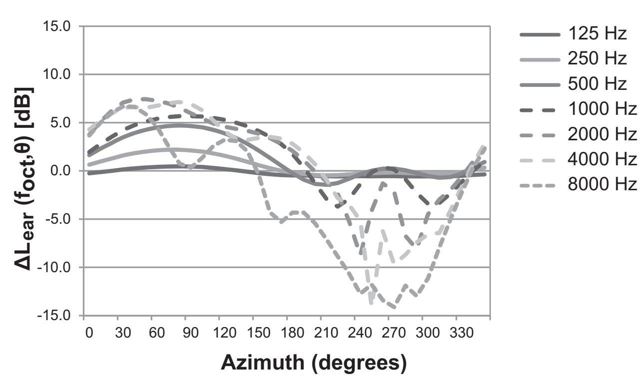

Figure 3. The transfer function of the receiver's head for the 0-degree elevation for the right ear, measured in an anechoic room using omnidirectional DPA 4060 miniature condenser microphones in front of the ear canal of a B&K Type 4128C head and torso simulator. The transfer functions of the receiver's head are calibrated using the sound level of the same microphones in the free field at the center head's position.

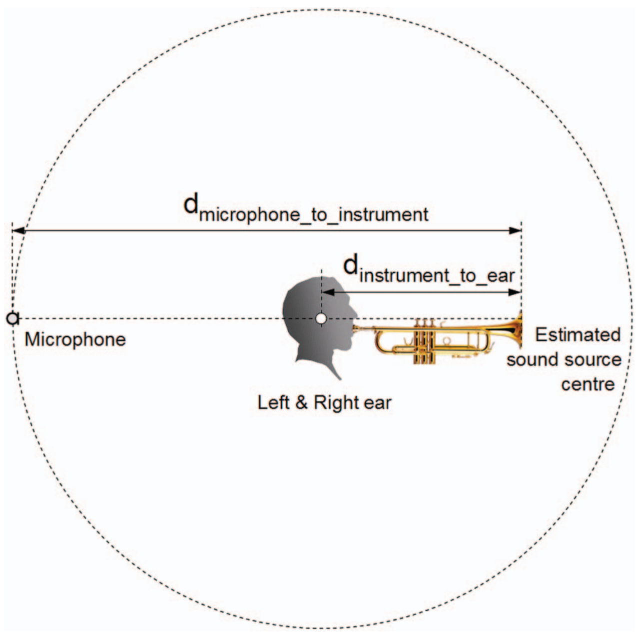

Figure 4. The direct sound of the own instrument is calculated at the ears by applying the inverse square law using the SPL measured on a line from the instrument through the ear to the measurement radius of the directivity measurement where the head is in the middle, see equation 3. See the online article for the color version of this figure.

sound level, because it was found that the agreement between measurement and model was best without HRTF.

It should be noted that bone-conducted sound might also contribute to the own instrument's sound level; however, this is not taken into account by the model.

Sound Power of Each Instrument

The sound power $L_w(f)$ is obtained from anechoic recordings of different musical fragments made available by Pätynen, Pulkki, and Lokki (2008) for each instrument. The equivalent SPL $L_{eq;front}(f)$ in dB is determined for every recording for the frontal microphone per instrument per musical piece. The silent parts in the recordings have not been removed, so results can be interpreted as a sound exposure level for the particular piece of music (although one could also use smaller time samples). An absolute sound level calibration was made available by Pätynen et al. The sound power $L_w(f)$ is calculated as

$$L_w(f) = L_{eq;front} + 20\,\mathrm{lg}(d) + 10\,\mathrm{lg}\left(\sum_{n=1}^{N} S_i 10^{\frac{L_I(f,\phi,\theta)}{10}}\right) \tag{4}$$

where $L_{eq;front}$ is the equivalent SPL in dB per frequency band in Hz in front of the musician (microphone no. 6, see Table 2 in Pätynen et al., 2008); S_i is the partial surface in m^2 per angle of the directivity data on a sphere of 1 m radius ($N = 614$); $L_I(f, \varphi, \theta)$ is the relative sound intensity at elevation φ and azimuth θ in degrees ($N = 614$), the musicians viewing direction is defined as 0 dB; and d is the microphone distance in meters ($d = 2.3$ m).

It should be noted that, by using a fixed sound power, the interaction between the acoustics and the musicians in the room is not taken into account.

Early and Late Reflected Sound

After the direct sound, sound arrives via reflections from the stage and hall boundaries. From laboratory studies using simulated sound fields, Gade (1989) concluded that reflections, arriving up until ~100 ms after the "orchestra onset," may support the musicians in playing ensemble. Wenmaekers, Hak, and van Luxemburg (2012) showed that, for an average size stage, first-order reflections indeed arrive within a maximum delay of 100 ms after the "orchestra onset." When a simultaneous onset of all instruments is assumed, the direct sound of the other player's instrument will arrive delayed, for instance, the sound traveling over 10 m distance will arrive 29 ms later. For the early reflections of this instrument to be "supportive," they should arrive within 100–29 = 71 ms after the arrival of the direct sound of that instrument. So, the time interval width between the arrival of the direct sound and the supportive early reflections depends on the source-to-receiver distance. To measure the amount of energy of this supportive early reflected sound, taking into account the distance-dependent time interval, the Early Support at various distances can be used, denoted $ST_{early;d}$, as introduced by Gade (1989) and modified by Wenmaekers et al. (2012), calculated as

$$ST_{early;d} = 10\,\mathrm{lg}\left(\frac{\int_{10}^{103-delay} p_d^2 dt}{\int_{0}^{10} p_{1m}^2 dt}\right) \tag{5}$$

where $ST_{early;d}$ is the Early Support at distance d in meters; p_d is the sound pressure measured at distance d; p_{1m} is the sound pressure measured at 1 m distance; and delay is the source-receiver distance divided by the speed of sound.

Research by Wenmaekers et al. (2012) has shown that in most cases the Early Support depends on the source-to-receiver distance. Using the Early Support parameter, the source-to-receiver distance d and the sound power of the sound source or instrument L_w, the early reflected sound level $L_{early\text{-}refl}$ is calculated as

$$L_{early-refl}(f, d) = L_w(f) + ST_{early;d}(f, d) - 11 \tag{6}$$

Late reflections describe the amount of reverberation. For the orchestra members, it is important to receive "feedback" from the hall to get an impression of what the audience is hearing (Gade, 1989). However, late reverberation can also mask the useful direct and early reflected sound. All reflections arriving 100 ms after the "orchestra onset" are considered late reflections. Again, the time interval start, relative to the arrival of the direct sound, is dependent on the distance. To measure the amount of energy of late reflected sound, the Late Support at various distances can be used, denoted $ST_{late;d}$, as introduced by Gade (1989) and modified by Wenmaekers et al. (2012), calculated as

$$ST_{late;d} = 10\,\mathrm{lg}\left(\frac{\int_{103-delay}^{\infty} p_d^2 dt}{\int_{0}^{10} p_{1m}^2 dt}\right) \tag{7}$$

In contrast to the $ST_{early;d}$, research by Wenmaekers et al. (2012) has shown that $ST_{late;d}$ is not dependent on the source-to-receiver distance, so a fixed value over the stage is valid (see section 5.5 in Wenmaekers et al., 2012). Similar to the early reflected sound level $L_{early\text{-}refl}$, the late reflected sound level $L_{late\text{-}refl}$ is determined from the sound power L_w of the instrument and the Late Support at various distances d denoted $ST_{late;d}$ as

$$L_{late-refl}(f) = L_w(f) + ST_{late;d}(f) - 11 \tag{8}$$

It has been suggested to use Sound Strength G as a parameter for describing the early and late reflected sound on stage (Dammerud et al., 2010), instead of using the ST parameters. Both types of parameters are defined in such a way, that the SPL of the impulse response (within a certain time interval) is compared with a reference SPL: the direct sound measured in the free field at a given distance. The reference SPL for G is determined at 10 m distance and for the ST parameters at 1 m distance. It should be noted that the reference level of the ST parameters is intended to be derived from a laboratory sound power measurement (Gade, 1992) and should not be derived from an in situ measurement on stage in which the floor reflection is present as is suggested by ISO 3382–1. Due to the difference in reference level, this results in a 20-dB difference between the parameters when the same time interval is used: for instance G_{10-100} and $ST_{early,d}$ at 1 m distance, both using a 10–100 ms time window. The 1-m reference, which is more or less equal to the distance between the own instrument and the ears, is conceptually clear for judging stage acoustic conditions and therefore the ST parameters are superior over G parameters which are using an arbitrary reference distance of 10 m. The choice of appropriate time intervals for judging early and late reflected sound levels, either using the ST or G definition, has been thoroughly investigated by the authors, see Wenmaekers et al. (2012). It was concluded that a time window defined relative to the departure of the direct sound, which has been introduced in the extended support parameters $ST_{early;d}$ and $ST_{late;d}$, is conceptually superior over a time window defined relative to the arrival of the direct sound at the receiver position, which is commonly used in G parameters like G_{late}.

The direction of arrival of the reflected sound might be important for both predicting sound levels and judging ensemble conditions. As explained in paragraph 1 of the "Model" section, varying (receiver) sensitivity with respect to the spatial sensitivity of the hearing system and the sound source directivity are not taken into account by the model for reflected sound. This means that in the prediction of the sound level of the reflected sound, the model assumes that the energy is transmitted and received equally spread over all directions. In theory, the "omnidirectional to omnidirectional transmission" might be valid under diffuse sound field conditions, which might be a valid approximation for the late reflected sound on a concert hall stage. The early reflected sound level depends more on discrete reflections and the "omnidirectional to omnidirectional transmission" might be less accurate. Especially the early reflections from the sidewalls of the stage might be reduced by the objects that have not been present during our measurements of $ST_{early;d}$ and $ST_{late;d}$. The possible impact of the orchestra members as objects on stage and the possible impact of directional reflections, to estimate the early and late reflected sound level, was not investigated in this article and is considered further research.

Measurement Conditions

The $ST_{early;d}$ and $ST_{late;d}$ are measured on a grid of measurement positions in the orchestra area, on stage, or in a rehearsal room. Omnidirectional transducers are used, preferably at 1 m height, kept at least 2 m away from any boundaries or objects. As suggested by ISO 3382–1, chairs and stands should be present to simulate the obstruction and scattering of sound by the orchestra (although no research has shown that this method does represent the orchestra obstruction sufficiently). To reduce the measurement uncertainty of single $ST_{early;d}$ and $ST_{late;d}$ measurements, it is recommended to use an average value over 5 stepwise rotations of the omnidirectional sound source (Hak et al., 2011) and to measure impulse responses with a decay range or impulse-to-noise ratio (Hak et al., 2012) of at least 45 dB. The reference level of ST, the direct sound at 1 m distance, is derived from a sound power measurement in a reverberation room, as suggested by Gade (1992). More background information on the measurement procedure can be found in Wenmaekers et al. (2012).

Input and Output

Early and Late Support for Various Halls

$ST_{early;d}$ and $ST_{late;d}$ have been measured by the authors in various concert halls (Wenmaekers et al., 2012), in orchestra pits (Wenmaekers & Hak, 2013), and in rehearsal rooms (Wenmaekers, Schmitz, & Hak, 2014). A logarithmic trend line can be calculated from various results of $ST_{early;d}$, per octave band for different source-to-receiver positions per hall. The trend line has the form "a lg (d) + b," where "a" and "b" are constants and "d" is the source-to-receiver distance. In Table 1, the 250 to 2000 Hz octave bands averaged constants to describe the $ST_{early;d}$ trend line, and the single number $ST_{late;d}$ are presented for a concert hall stage, an orchestra pit, and a rehearsal room. Also, the average reverberation time T_{mid} (500 and 1000 Hz) is given. In Figure 5, the trend lines for $ST_{early;d}$ are presented. For the orchestra pit, three different trends are given, each having a distinctive shape for different types of source and receiver positions: both positions in the open part; both positions in the covered part; and just one of both positions in the open or covered part (Wenmaekers & Hak, 2013). In the calculations that will be presented in section "Case study," the single number values for $\mathrm{ST_{early;d}}$ and $\mathrm{ST_{late;d}}$ have been used for each individual octave band.

Orchestral Layout

Based on the typical Mahler Symphony 1 orchestration and the typical American orchestra layout (Meyer, 2009), an orchestra setup is selected for the model with all musicians positioned on a rectangular grid, see Figure 6. The receiving musician 8 (violin) is used in this article as an example. In case of the concert hall stage, musicians 56–58 and 63–74 are elevated by 0.3 m and musicians 59–62 and 81–82 are elevated by 0.6 m to simulate risers. In the case of an orchestra pit, it is assumed that all woodwind, brass, horn, and percussion players are positioned in the covered part of the pit and all strings sections are positioned in the open part of the pit. A distance of 1.3 m (side to side) and 1.6 m (back to back) was assumed between the musicians on the grid, resulting in a 2.1-m^2 area per musician

Table 1
Early Support $ST_{early;d}$, Late Support $ST_{late;d}$, and Reverberation Time T_{Mid} Measured in a Concert Hall, Orchestra Pit, and Rehearsal Room

	$ST_{\mathrm{early;d}}$ = a lg (d) + b [dB]		$ST_{\mathrm{late;d}}$ [dB]	T_{mid} [s]
Hall	a	b	Average	Average
Concert hall stage 14,400 m^3	−1.6	−11.0	−15.2	2.0
Rehearsal room 2,500 m^3	−2.8	−9.6	−11.8	1.2
Orchestra pit (Covered-Covered)	−7.0	−1.5	−15.2	1.3
Orchestra pit (Open-Covered)	−13.4	+.4	−17.3	1.3
Orchestra pit (Open-Open)	−5.2	−9.3	−16.0	1.3

(similar to Dammerud and Barron (2010) who used 2.3 m^2 per musician).

Sound Power of Musical Pieces

To asses spectral and level differences between instruments, the sound power L_w is needed, which is derived from the calibrated anechoic recordings of musical pieces by Pätynen et al. (2008). Separate instrument recordings were made of different orchestral excerpts of music. From the recordings of the Mahler Symphony no. 1 sample (first 2:12 min of the fourth part) and Bruckner Symphony no. 8 sample (1:27 min), the equivalent sound power levels have been determined using equation 4. Figure 7 shows the A-weighted sound power level per instrument per musical piece, as an equivalent sound power level over the musical sample. Large differences between the two pieces occur only at the violin sections. Because of relatively small differences between the two pieces and because only the Mahler piece has a percussion part, Mahler will be used for calculations in this article.

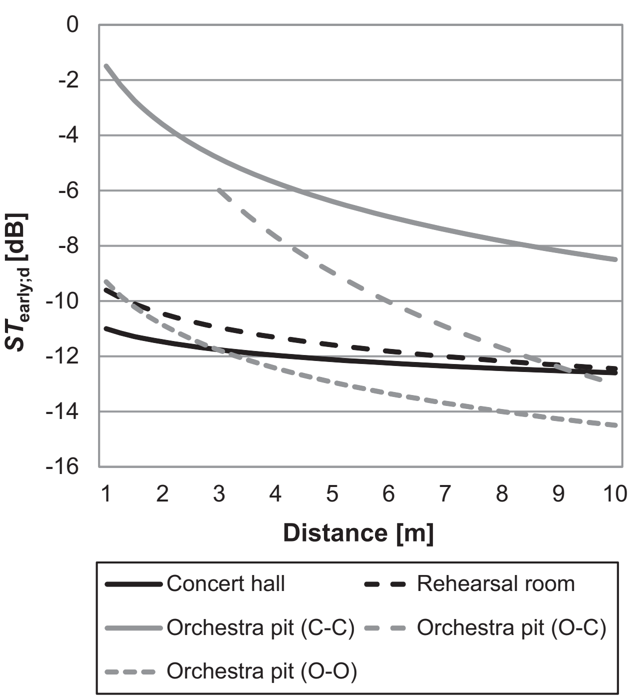

Figure 5. Trend lines of $ST_{\mathrm{early;d}}$, as a function of distance measured in a concert hall (R^2 = 0.04, this typical concert hall stage has almost no decrease of $ST_{\mathrm{early;d}}$ over distance), rehearsal room (R^2 = 0.45), and orchestra pit: O-O, both positions in the open part (R^2 = 0.74); C-C, both positions in the covered part (R^2 = 0.88); and O-C, just one of both positions in the open or covered part (R^2 = 0.85).

Output

The output of the model gives values for $L_{direct}(f)$, $L_{early\text{-}refl}(f)$, and $L_{late\text{-}refl}(f)$ for every combination of source and receiver position. Also, values for $L_{total}(f)$ can be determined, which is the energetic summation of $L_{direct}(f)$, $L_{early\text{-}refl}(f)$, and $L_{late\text{-}refl}(f)$. For a symphony orchestra of 80 musicians, this results in 80 × 80 = 6,400 values for every parameter and frequency band. The sound energy of sound sources with equal instrument and musical parts are energetically summed to study grouping effects.

Human ears are highly sophisticated sound receivers, with varying sensitivity with respect to frequency. The varying sensitivity is introduced in this model by A or C weighting the sound level in accordance with IEC 61672. The A-weighting is commonly used when assessing relatively high sound exposure levels, even though it represents the low isophone curve with 40 dB at 1000 Hz. The C-weighting, representing the isophone curve with 100 dB at 1000 Hz, is more appropriate when assessing loudness in symphony orchestras, as sound levels are often 80 dB up to 100 dB. Besides

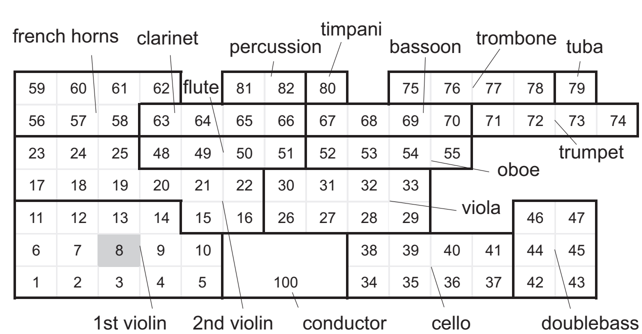

Figure 6. Generic orchestra setup for Mahler Symphony 1 (receiver 8 used in the Case Study section is marked grey) Strings: 1–14: 1st violin, 15–25: 2nd violin, 26–33: viola, 34–41: violoncello, 42–47: double bass. Woodwinds: 48–51: flute, 52–55: oboe, 63–66: clarinet, 67–70: bassoon. Brass: 71–74: trumpet, 75–78: trombone, 79: tuba. Separate instruments: 56–62: French horn, 80: timpani, 81–82: percussion.

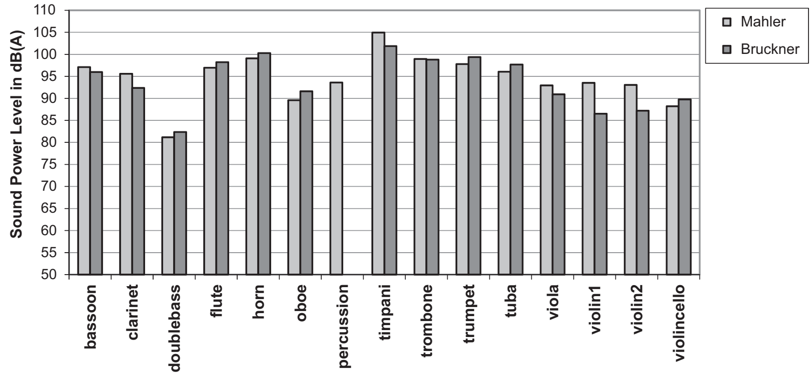

Figure 7. Average A-weighted sound power level per instrument for different musical pieces Mahler Symphony no. 1 sample (2:12 min) and Bruckner Symphony no. 8 sample (1:27 min).

weighted levels, the model could also be used to consider separate octave bands 125 to 8000 Hz.

Validation

Measurement Setup

The direct binaural sound level of the own instrument has been measured in an anechoic room with two DPA 4060 miniature condenser microphones fixed in front of the musician's ears, see Figure 8. Also, a B&K Type 4189-A-021 microphone was positioned at 2 m distance from the musicians' ears at equal height, see Figure 8, denoted Ref. This microphone is used to determine the reference sound level in front of the musician $L_{eq;microphone}$ (*f*), see equation 3. The SPLs measured using the DPA microphones were corrected to match the flat frequency response of the B&K microphone, based on a comparison study of the microphones in a diffuse field (reverberation room) and direct field (anechoic room). However, it should be noted that at high frequencies for both microphones, the sound levels measured close to the ear are 2.5, 4.5, and 2.0 dB higher in the octave bands 2, 4, and 8 kHz, respectively, than in the free field. These differences are considered to be caused by the sound field, and not by the type of microphone, so no additional correction is made.

Procedure

Various musical instruments were investigated in the research: flute (2x), piccolo (2x), trumpet, flugelhorn, bass trombone, trombone, and violin. Every musician was asked to play C major scales in the native playing range of the instrument over two octaves up and down, with altered articulation (staccato and legato) and musical dynamics (piano and forte). All tones were played with constant speed. While playing, calibrated recordings have been made using the three microphones. The average SPL was determined for the whole recording session. Afterward, the background noise level of the measurement system was determined. In this research, sound levels are only presented if they are at least 10 dB above the background noise level.

Distance Between Instrument and Ears

Part of the model's input in equation 3 are the geometrical parameters elevation φ, azimuth θ, and distance between the instrument and the musicians left and right ear. The applied angles of elevation φ and azimuth θ are illustrated in Figure 9. The values determined for the musicians in this research are presented in Table 2. For the flute, the geometrical parameters were determined relative to half of the tube length at 40 cm to the left ear and 20 cm to the right ear and, for the piccolo, at 28 cm to the left ear and 15 cm to the right ear. For the trombone player's right ear, the geometrical parameters were deter-

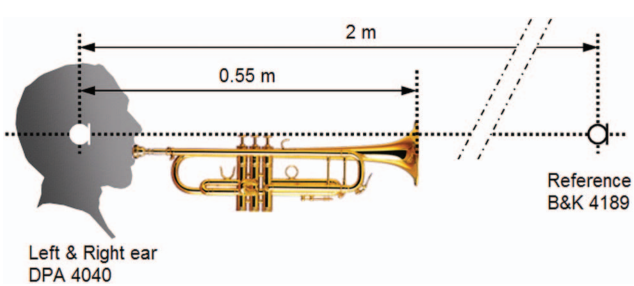

Figure 8. Side view of the microphone setup. Two DPA 4040 microphones are positioned in front of the musician's ears. The distance from the center of the musician's head to the reference microphone B&K 4189 is 2 m. The distance between the center of the sound source (the bell of the trumpet) to the musician's ears is 0.55 m. See the online article for the color version of this figure.

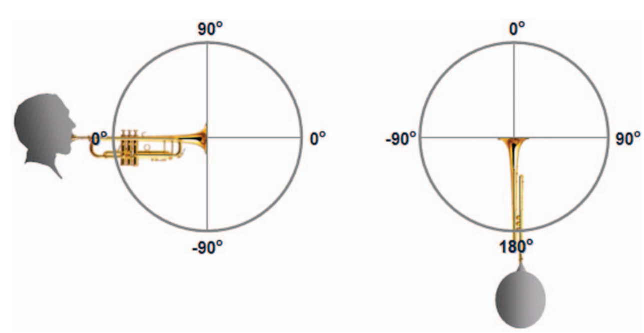

Figure 9. Side view showing Elevation θ (left) and top view showing Azimuth φ (right) used to describe the angle between the sound source center (in this example the bell of the trumpet) and the musicians' ears. See the online article for the color version of this figure.

Table 2
Geometrical Parameters for the Angle and Distance Between Instrument and Ears

Instrument	Elevation θ [°]	Azimuth φ [°]	D1[a] [m]	D2[b] [m]
Flute (L)	10	260	0.30	0.40
Flute (R)	10	250	0.30	0.20
Piccolo (L)	10	260	0.16	0.28
Piccolo (R)	10	250	0.16	0.15
Trumpet/Flugelhorn (L&R)	0	180	0.55	0.55
(Bass) Trombone (L)	10	180	0.40	0.30
(Bass) Trombone (R)	10	150	0.40	0.50
Violin (L)	30	180	0.20	0.20
Violin (R)	30	135	0.20	0.25

Note. L = left; R = right. See Figure 4 for an graphical explanation of the geometry.
[a] Distance between middle of the head and the reference point on the musical instrument. [b] Distance between each ear and the reference point on the musical instrument.

mined relative to the bell, slightly on the left at 50 cm. The trombone player's left ear is in proximity to the tubes on the shoulder, which also radiates sound, so the geometrical parameters were determined relative to 30 cm. For both ears of the trumpet/flugelhorn player, the geometrical parameters were determined relative to the bell in front of the player at 55 cm. For the violin, the geometrical parameters were determined relative to the bridge, more or less in the middle of the soundboard at 20 cm to the left ear and 25 cm to the right ear. The neck of the violin was pointing toward 330 degrees azimuth.

Results From Binaural Measurements

Table 3 shows the measured level difference in dB between the left and right ear per instrument. The results are presented per octave band and A-weighted (broadband). For reference, the absolute A-weighted sound level is also presented, which shows that the direct sound of the sound instrument is above 90 dB(A) in most cases, and even up to 100 dB(A) in one case (remember that this is averaged over playing scales both in Piano and Forte). The A-weighted sound level difference at the two ears varies from −3.4 to −7.4 dB for the flutes and piccolos positioned on the right side of the head. For the trumpet and flugelhorn, a +0.7 dB and −1.7 dB A-weighted difference is found, respectively, caused by the bells being slightly off center to the left for the trumpet and off center to the right for the flugelhorn. A striking +4.7 and +4.9 dB A-weighted level difference is found for the trombones, with differences of +11 to +14 in the high frequency bands. For the violin, an A-weighted level difference is found of +2.3 dB, which was expected to be (much) higher.

Interaural level differences (ILD) have been reported earlier by Meyer (1981), measured in an anechoic room, and Schmidt (2011), measured in a rehearsal room. Schmidt reported a level difference of −7.4 dB for the flute and −6.7 for the piccolo. For the trumpet, values of 0 dB and +1.4 dB were reported by Meyer and Schmidt, respectively, and for the trombone +3 dB and +3.8 dB. These values are (more or less) similar to what was found in this study. It is striking though, that Meyer and Schmidt found a level difference for the violin of +10 dB and +5.3 dB, respectively, which is much higher than the +2.3 dB that was found in this study. However, it should be noted that the playing style of the violin player might have a large influence on the ILD. Using equation 3, we estimated that the ILD is +2.3 dB when the neck of the violin is pointing toward 330 degrees azimuth and the ILD is estimated to be +8 dB when the neck of the violin is pointing toward 270 degrees azimuth. This might explain the differences between the various studies.

Model Calculations of Levels From the Musician's Own Instrument

Using the sound level measured in front of the musician at 2 m distance, the directivity per angle from Pätynen & Lokki, and the geometrical parameters as presented in Table 2, the sound level at the musicians' ears have been estimated by equation 3. The results are

Table 3
Measured Interaural Level Difference per Instrument, Measured in an Anechoic Room

Instrument	125 L-R	250 L-R	500 L-R	1000 L-R	2000 L-R	4000 L-R	8000 L-R	A-weighted L	A-weighted R	A-weighted L-R
Flute 1		−2	−6	−7	−1	+1		86	93	−6.4
Piccolo 1			−4	−10	−1	−9	−6	90	93	−3.4
Flute 2		−3	−6	−8	−4	−3		86	93	−7.4
Piccolo 2			−5	−13	−2	−8	−9	92	97	−4.3
Trumpet	+1	+1	+1	−1	+3	+3	+4	97	96	+0.7
Flugelhorn	0	0	0	−2	−1	−2		98	100	−1.7
Bass trombone	+3	+3	+4	+6	+13	+12		96	91	+4.9
Trombone	+2	+3	+3	+6	+11	+14		97	92	+4.7
Violin	−1	0	+1	+4	0	0	+4	92	90	+2.3

Note. L = left; R = right.

Table 4
The Difference in Interaural Level Difference Between Measured and Estimated, Left Ear

Instrument	125 L	250 L	500 L	1000 L	2000 L	4000 L	8000 L	A-weighted Meas	A-weighted Estim	A-weighted M-E
Flute 1		3	9	−2	−1	0		86	87	−0.7
Piccolo 1			−1	−2	3	−7	−5	90	88	1.7
Flute 2		6	7	−4	−5	−1		86	89	−3.7
Piccolo 2			0	−2	4	−4	−5	92	89	3.3
Trumpet	0	−1	3	−6	3	2	0	97	99	−2.2
Flugelhorn	0	0	4	−3	0	0		98	100	−2.1
Bass trombone	−3	−4	−1	−2	4	−1		96	97	−1.4
Trombone	−3	−2	−1	0	5	2		97	97	−0.4
Violin		2	8	−2	−1	−1	8	92	93	−0.9

Note. L = left; Meas = measured; Estim = estimated; M = measured; E = estimated.

compared to the actually measured sound levels. The difference between the measured and estimated values are presented in Table 4 for the left ear and Table 5 for the right ear. The differences are presented per octave band and A-weighted. For reference, the absolute A-weighted sound level is also presented. In the column to the right, the A-weighted difference between measured and estimated is presented.

The comparison of the measured and estimated binaural sound levels shows that, for the individual frequency bands, errors are found up to 9 dB. The mean absolute error is ~3 dB for the individual frequency bands. For most instruments, both positive and negative errors occur over the frequency range, except for the trombones at the right ear that show only negative errors. When looking at the A-weighted errors, for the left ear, the model overestimates the A-weighted level by 0.7 to 3.7 dB. For the right ear, the model overestimates the A-weighted level by values between 0.1 and 2.8 dB. Exceptions are the piccolos that are underestimated by 1.7 to 3.3 dB.

Conclusions on the Validation Study

Looking at the individual frequency bands, we can conclude that the model is accurate within a mean absolute deviation of 3 dB, with maxima up to 9 dB. This relatively large deviation can be caused by the fact that the musician's ear is so close to the musical instrument, that the instrument cannot be considered as a point source. Also, the directivity that is used in the model was obtained from different instruments playing a different repertoire. Additional measurements with multiple musicians and instruments could produce more uniform results, on the other hand. However, considering the model's purpose being to investigate the different contributions of many different aspects to the noise exposure of musicians within an orchestra, we can conclude that the prediction of the direct A-weighted sound exposure of the own instrument at the musicians ear can be done within 1.7 dB(A) average deviation with a maximum deviation of 3.7 dB(A).

Case Study

Using the symphony orchestra sound level prediction model, as presented in the "Model" section, and the input data, as presented in section "Input and Output," a calculation has been made for the violin position 8 (see Figure 10) as an example. For every instrument group, the direct, early reflected, and late reflected sound level as received at the violin player's left ear, and the total sound level received per group at both ears, is calculated. It should be noted that the equivalent sound level is calculated over the whole musical sample, resulting in a time-average balance. From a musical point of view, this may only show an average sound level balance, not the actual balance in a certain shorter piece like a note, measure, or phrase.

In Figure 10, the results for the violin's position calculation are presented in four bar graphs, the first graphs "a" and "b" both

Table 5
The Difference in Interaural Level Difference Between Measured and Estimated, Right Ear

Instrument	125 R	250 R	500 R	1000 R	2000 R	4000 R	8000 R	A-weighted Meas	A-weighted Estim	A-weighted M-E
Flute 1		−1	9	0	−5	−6		93	93	−0.1
Piccolo 1			−3	2	−1	−3	−4	93	93	−0.3
Flute 2		2	6	−3	−7	−3		93	95	−2.2
Piccolo 2			0	5	1	−2	−2	97	94	2.2
Trumpet	−1	−2	2	−4	0	−1	−3	96	99	−2.8
Flugelhorn	0	0	5	−1	0	3		100	100	−0.5
Bass trombone	−2	−3	−2	−4	−5	−8		91	93	−2.7
Trombone	−1	−2	−1	−3	−2	−6		92	94	−1.5
Violin		4	7	−4	1	3	6	90	91	−1.0

Note. R = right; Meas = measured; Estim = estimated; M = measured; E = estimated.

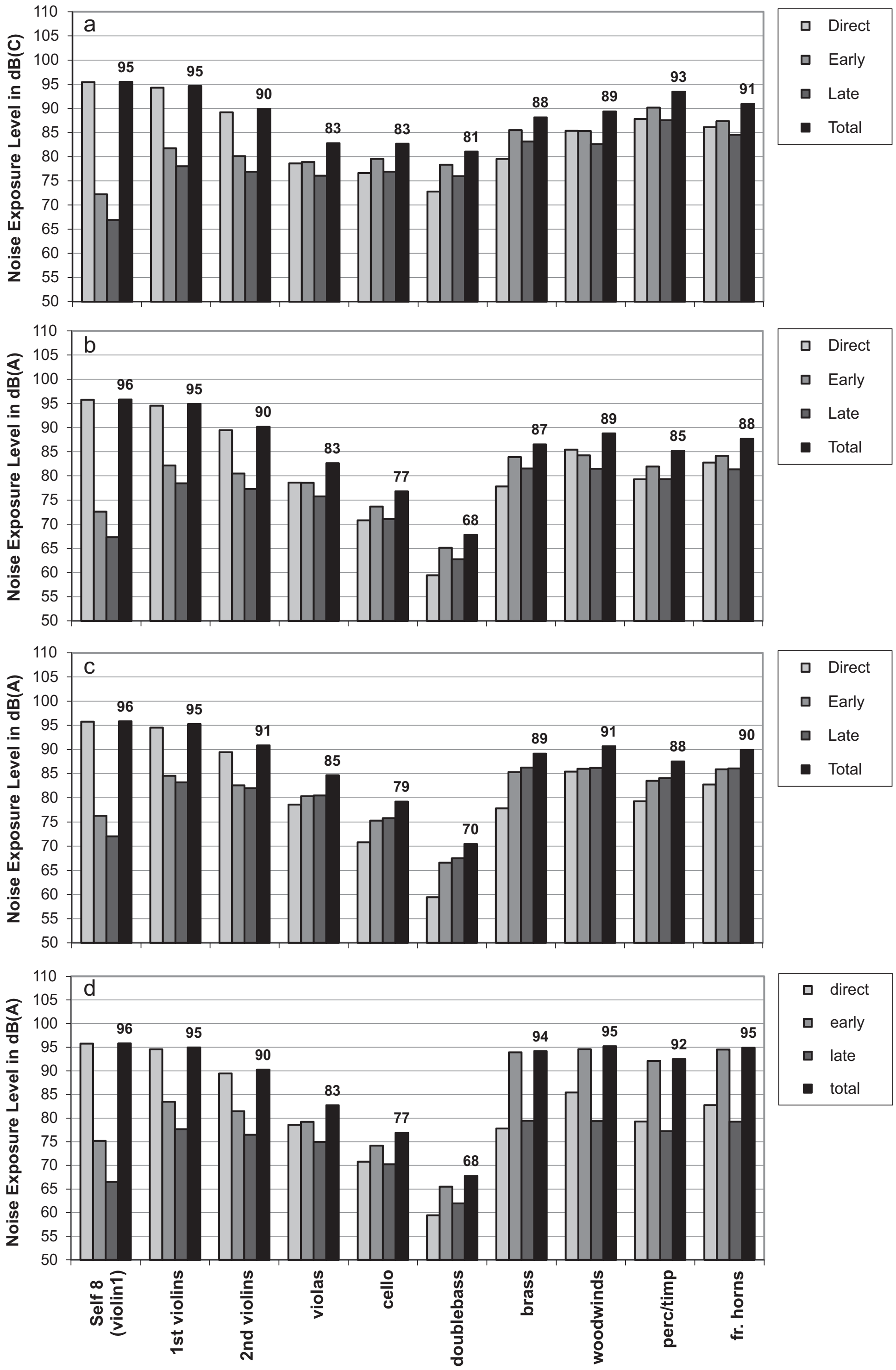

Figure 10. Sound level balance in dB for the violin position 8 (left ear) per instrument group, for the direct sound (direct), early reflected sound (early), late reflected sound (late), and the summation of direct, early, and late (total). (a) Concert Hall, C-weighted; (b) Concert Hall, A-weighted; (c) Rehearsal Room, A-weighted; (d) Orchestra Pit, A-weighted.

represent the sound levels for the concert hall stage, where graph "a" shows C-weighted values and graph "b" shows A-weighted values. It is shown that the C-weighted levels are 3 to 13 dB higher than the A-weighted levels for the instrument groups with more low frequency energy: cello, double bass, timpani, and horns. For the instrument groups nearby position 8, the 1st and 2nd violins, the direct sound level dominates over the reflected sound level. For instruments sitting further away, the contributions of direct, early and late sound are within a range of 5 dB and reflected sound tends to dominate over direct sound. The total A-weighted levels, commonly used for judging noise exposure, show that the direct sound of the own instrument and the own group are more or less equal, with 96 and 95 dB(A), respectively. All other instrument groups, except for the cello and double bass, are 5 to 12 dB(A) lower in total level, thus contributing moderately to the noise exposure at the violin's position.

The 2nd to 4th bar graph in Figure 10, graphs "b," "c," and "d," show a comparison of A-weighted sound levels for the concert hall (b), rehearsal room (c), and orchestra pit (d), see Table 1 for their properties. In this case, only the commonly used A-weighting is chosen for presentation to be able to evaluate the noise exposure. The late reflected sound level is 3.4 dB louder in the rehearsal room for all instruments, resulting in ~2 dB higher total levels from instruments sitting further away. The noise exposure at the violins position, caused by all instruments, increases by 0.8 dB(A). In the orchestra pit, a dramatic increase is shown for the early reflected sound coming from the players in the covered part of the pit causing an increase in total level up to 5 dB(A). (This effect might be somewhat exaggerated because the orchestra pit was empty during the measurement of the $ST_{early;d}$ and $ST_{late;d}$ parameters; and because an omnidirectional source was used). The late reflected sound level is (much) lower.

The total sound level (self and all others) at the violin position is 100.1 dB(A) for the concert hall, 100.8 dB(A) for the rehearsal room, and 102.8 dB(A) in the orchestra pit. A difference of almost 3 dB between on stage and orchestra pit conditions is calculated, meaning a doubling of the noise dose.

In Figure 11, the sound level in dB(A) is presented as an average within each instrument group. The total sound level of the own instrument within the group is shown together with the direct, early, late, and total sound received from all others. In general, sound levels are relatively high for this particular musical piece. Total equivalent sound levels are calculated ranging from 94 to 101 dB(A), even if one's own instrument is not producing levels above 85 dB(A), like the cello, double bass, and the conductor. For the high strings, the own instrument sound is equally loud as the direct sound of all others. For the brass, woodwinds, timpani, and French horns, the direct sound of all others appears to be the loudest component. The differences between the right and left ear are small, 1 to 2 dB(A). Because the violin players direct their left ear toward the orchestra, direct sound from all others is 1.5 dB(A) higher at the left ear. In a similar way, for the cello and double bass, the right ear receives up to 4 dB(A) more direct sound from others.

In accordance with European Directive, 2003/10/EC (Directive, 2003), one is required to wear hearing protection above equivalent

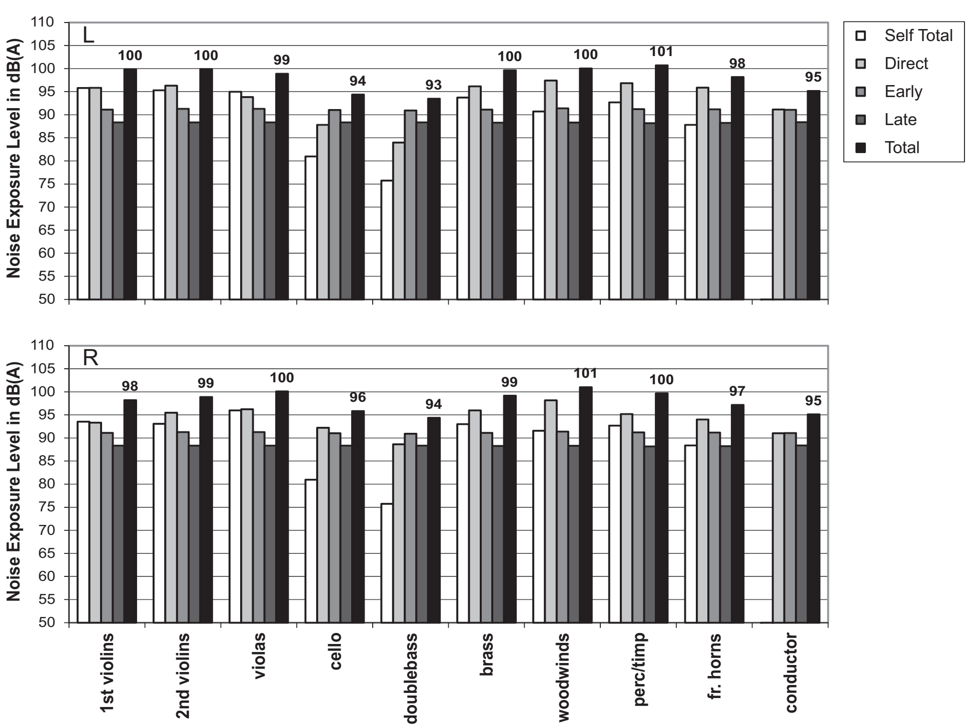

Figure 11. Average sound levels in dB(A) per instrument group on a concert hall stage for the left ear (L) and right ear (R), for the total own instrument sound (Self Total), the direct sound of all others (Direct), early reflected sound of all others (Early), late reflected sound of all others (Late), and the summation of self, and direct, early, and late of all others (Total).

sound levels $L_{A;eq} \geq 85$ dB(A). The measured results from the individual players in the anechoic room, as presented in the "Validation" section, and the calculated results for the full orchestra, as presented in this section, show equivalent sound levels well above 85 dB(A), up to 100 dB(A). Peak levels could not be assessed using the model. Due to the large dynamics of music, the noise exposure should be evaluated over a larger period than one musical sample of 2 min. Nevertheless, if one would like to keep the daily noise dose below 85 dB(A), and the $L_{A;eq}$ is 100 dB(A) for this particular Mahler Symphony no. 1 sample (2:12 min), one should rehearse this excerpt not more than seven times a day without wearing ear plugs, while remaining practically silent during the rest of the day. This example illustrates that noise exposure control for musicians can be quite a challenge.

Discussion

The presented work is a result of an ongoing study on the development of a model to estimate the sound levels within an orchestra. It is shown that the model has much potential for studying the influence of architectural as well as acoustical aspects on the sound levels that occur in a symphony orchestra. It is also important to mention the limitations of this model:

- The directivity of the instruments is not taken into account in the (measured) early and late reflected sound level. In a similar way, the directivity of the listener's ears has not been taken into account.
- The attenuation by the orchestra is not taken into account in the measured room acoustical parameters, which may result in an overestimation of the early and late reflected sound.
- The outcome of the model could not be validated based on actual sound level measurements.

In spite of these shortcomings, it is shown that the model has the potential to give valuable insights in the sound level distribution of different instruments in a symphonic orchestra. The study of the sound levels has shown to be interesting from a health point of view. To effectively control sound exposure of musicians in an orchestra, it is important to understand the contributions of the various aspects that determine the total exposure. The introduced prediction model can give insight into all these aspects, provided that the uncertainties are further reduced. In the future, it would be interesting to use the model to study the impact of screens between musicians and different orchestra setups on sound exposure. Also, other types of stage environments could be analyzed and the impact of architectural elements on within-orchestra sound levels could be investigated.

The study of these sound levels has also shown to be interesting from a musical point of view. The results might be useful to study the effect of orchestra setup together with room acoustics on ensemble playing for different pieces of music. For instance, it would be interesting to investigate whether the predicted sound level balance matches the perceived loudness balance by orchestral musicians under various conditions.

Future work on the development of the model should focus on

- Measuring or calculating impulse responses with source and receiver directivity.
- Finding the impact of the orchestra itself on the early and late reflected sound.
- Validating the predictions of the model by (total) sound level measurements under different acoustic conditions.

Conclusion

In this article, a model is presented to study the direct, early reflected, and late reflected sound distribution in symphony orchestras. The model for the direct sound level of the own instrument has been validated by measurements, showing an average deviation of 1.7 dB(A). The results from the measurements and model revealed that, in some cases, like the violin player, the positioning of the instrument by the musician can have a large influence on the interaural-level differences. Interaural-level differences reach up to 13 dB for separate octave bands and 7.4 dB for A-weighted sound levels. This confirms that when judging musicians' sound exposure, the sound levels should be studied close to each ear separately, instead of only using one position on the musician's shoulder as suggested by ISO 9612.

The distribution of the direct, early reflected, and late reflected sound levels have been studied for a concert hall stage, a rehearsal room, and an orchestra pit using the model. Results indicate that the contribution of each aspect (own direct sound, others' direct sound, early and late reflected sound) to the total sound level at the receiving musician's position can be in the same order of magnitude. This shows that actual sound level measurements in an orchestra could never reveal the impact of the various aspects individually, and a model, as presented in this article, is indispensable.

As an example, the sound level distribution has been investigated for a sample of music from Mahler Symphony no. 1 at a first violin's position in the orchestra. The balance between direct and reflected sound depends on the distance to the various instrument groups. The difference in acoustics of the concert hall stage and the rehearsal room is expressed by the increase in the late reflected sound level in the rehearsal room by 3.4 dB, which results in a 0.8 dB(A) higher calculated total sound exposure at the violin's position. The difference in acoustics of the concert hall and the orchestra pit is expressed by the increase in the early reflected sound level, especially for the instruments positioned in the covered part of the pit. At the violin's position, a difference in total sound level of 2.7 dB(A) between on stage and orchestra pit conditions is calculated, meaning a doubling of the noise dose.

The results from the example show that the model has potential for studying the influence of architectural as well as acoustical aspects on the sound levels that occur in a symphonic orchestra.

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